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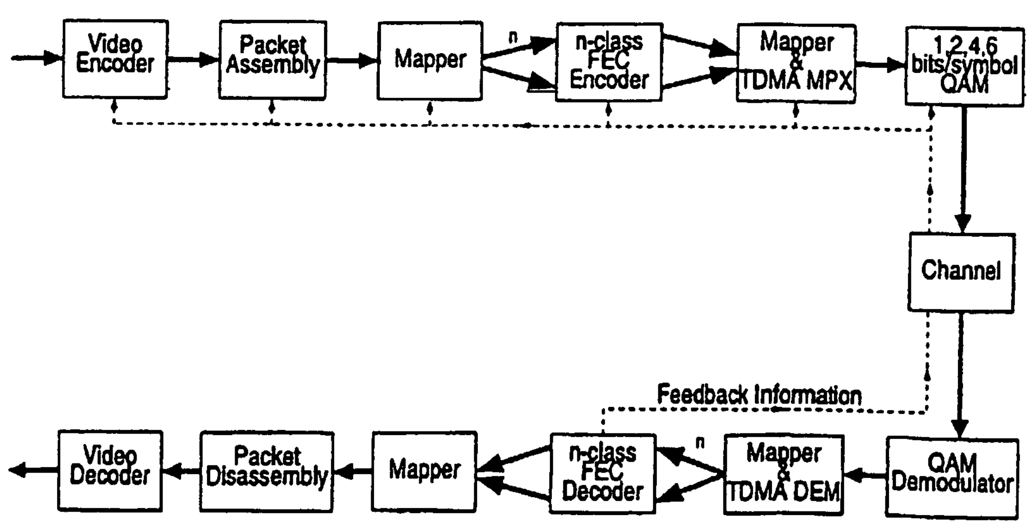
(54) Title: ADAPTIVE JOINT-DETECTION CDMA VIDEO TRANSCEIVER



(57) Abstract: In a near-instantaneously adaptive joint-detection CDMA-based transceiver used for wireless video telephony a method for transmission of a multimedia signal is described, the method comprising: providing a transmitter operable to transmit in a plurality of modulation modes varying in bit rate and error resilience between a highest bit rate, lowest error resilience mode and a lowest bit rate, highest error resilience mode; obtaining a channel quality measure for current transmission; and switching to a more or less error resilient modulation mode each time the channel quality measure respectively degrades or improves by a defined amount, whereby multimedia signal quality varies smoothly with varying channel quality of the transmission medium.

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ADAPTIVE JOINT-DETECTION COMA VIDEO TRANSCEIVER

1 Background of the Invention

The invention relates to burst-by-burst adaptive joint-detection Code Division Multiple Access (CDMA) based transmission of multimedia signals, such as interactive video or audio, speech etc.

In contrast to the *burst-by-burst reconfigurable CDMA* multimedia transceivers described in this document, the term *statically reconfigurable* found in this context in the literature refers to multimedia transceivers that cannot be near-instantaneously reconfigured. More explicitly, the previously proposed *statically reconfigurable* video transceivers were reconfigured on a long-term basis under the base station's control, invoking for example in the central cell region - where benign channel conditions prevail - a less robust, but high-throughput modulation mode, such as 4 bit/symbol Quadrature Amplitude Modulation (16QAM), which was capable of transmitting a quadruple number of bits and hence ensured a better video quality. By contrast, a robust, but low-throughput modulation mode, such as 1 bit/symbol Binary Phase Shift Keying (BPSK) can be employed near the edge of the propagation cell, where hostile propagation conditions prevail. This prevented a premature hand-over at the cost of a reduced video quality.

The philosophy of the fixed, but programable-rate proprietary video codecs and statically reconfigurable multi-mode video transceivers presented by Streit *et al.* for example in References [1] was that irrespective of the video motion activity experienced, the specially designed video codecs generated a constant number of bits per video frame. For example, for videophony over the second-generation Global System of Mobile Communications known as the GSM system at 13 kbps and assuming a video scanning rate of 10 frames/s, 1300 bits per video frame have to be generated. Specifically, two families of video codecs were designed, one refraining from using error-sensitive run-length coding techniques and exhibiting the highest possible error resilience and another, aiming for the highest possible compression ratio. This fixed-rate approach had the advantage of requiring no adaptive feedback controlled bitrate fluctuation smoothing buffering and hence exhibited no objectionable video latency or delay. Furthermore, these video codecs were amenable to video telephony over fixed-rate second-generation mobile radio systems, such as the GSM.

The fixed bitrate of the above proprietary video codecs is in contrast to existing standard video codecs,

such as the Motion Pictures Expert Group codecs known as MPEG1 and MPEG2 or the ITU's H.263 codec, where the time-variant video motion activity and the variable-length coding techniques employed result in a time-variant bitrate fluctuation and a near-constant perceptual video quality. This time-variant bitrate fluctuation can be mitigated by employing adaptive feed-back controlled buffering, which potentially increases the latency or delay of the codec and hence it is often objectionable for example in interactive videophony. The schemes presented by Streit *et al.* in References [1] result in slightly variable video quality at a constant bitrate, while refraining from employing buffering, which again, would result in latency in interactive videophony. A range of techniques, which can be invoked, in order to render the family of variable-length coded, highly bandwidth-efficient, but potentially error-sensitive class of standard video codecs, such as the H.263 arrangement, amenable to error-resilient, low-latency interactive wireless multimode videophony was summarised in [2]. The adaptive video rate control and packetisation algorithm of [2] generates the required number of bits for the burst-by-burst adaptive transceiver, depending the on the capacity of the current packet, as determined by the current modem mode. Further error-resilient H.263-based schemes were contrived for example by Färber, Steinbach and Girod at Erlangen University [3], while Sadka, Eryurtlu and Kondoz [4] from Surrey University proposed a range of improvements to the H.263 scheme. Following the above portrayal of the prior art in both video compression and statically reconfigurable narroband modulation, let us now consider the philosophy of wideband burst-by-burst adaptive quadrature amplitude modulation (AQAM) in more depth.

In burst-by-burst adaptive modulation a higher-order modulation scheme is invoked, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, a more robust lower order modulation scheme is employed, when the channel exhibits inferior channel quality, in order to improve the mean Bit Error Ratio (BER) performance. A practical scenario, where adaptive modulation can be applied is, when a reliable, low-delay feedback path is created between the transmitter and receiver, for example by superimposing the estimated channel quality perceived by the receiver on the reverse-direction messages of a duplex interactive channel. The transmitter then adjusts its modem mode according to this perceived channel quality.

Recent developments in adaptive modulation over a narrow-band channel environment have been pioneered by Webb and Steele [5], where the modulation adaptation was utilized in a Digital European Cordless Telephone - like (DECT) system. The concept of variable rate adaptive modulation was also advanced by Sampei *et al* [6], showing promising advantages, when compared to fixed modulation in terms of spectral efficiency, BER performance and robustness against channel delay spread. In another paper, the numerical upper bound performance of adaptive modulation in a slow Rayleigh flat-fading channel was evaluated by Torrance *et al*[7] and subsequently, the optimization of the switching threshold

levels using Powell minimization was used in order to achieve a targeted performance [8, 9]. In addition, adaptive modulation was also studied in conjunction with channel coding and power control techniques by Matsuoka *et al* [6] as well as Goldsmith *et al.*[10].

In the narrow-band channel environment, the quality of the channel was determined by the short term Signal to Noise Ratio (SNR) of the received burst, which was then used as a criterion in order to choose the appropriate modulation mode for the transmitter, based on a list of switching threshold levels, l_n [5, 9]. However, in a wideband environment, this criterion is not an accurate measure for judging the quality of the channel, where the existence of multi-path components produces not only power attenuation of the transmission burst, but also intersymbol interference. Subsequently, a new criterion has to be defined to estimate the wideband channel quality in order to choose the appropriate modulation scheme.

2 Summary of the Invention

Particular and preferred aspects of the invention are set out in the accompanying independent and dependent claims. Features of the dependent claims may be combined with those of the independent claims as appropriate and used in combinations other than those explicitly set out in the claims.

The performance benefits of burst-by-burst adaptive modulation assisted CDMA are described, employing a higher-order modulation mode in transmission bursts, when the instantaneous channel quality is favourable, ie when the received signal is unimpaired by co-channel interferers. This procedure is employed, in order to increase the system's bits per symbol (BPS) capacity and conversely, invoking a more robust, lower order modulation mode, when the channel exhibits inferior channel quality. Therefore the associated bit rate will be time-variant.

It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source-signal representation quality - such as video or audio quality - versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advantage of the described technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This can achieved on a near-instantaneous basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when a mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while

in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

3 Brief Description of the Drawings

For a better understanding of the invention and to show how the same may be carried into effect reference is now made by way of example to the accompanying drawings, in which:

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3.1 State-of-the-art

Burst-by-burst adaptive quadrature amplitude modulation (AQAM) was contrived by Steele and Webb [5], in order for the transceiver to cope with the time-variant channel quality of narrowband fading channels. Further related research was conducted at the University of Osaka by Sampei and his colleagues, investigating variable coding rate concatenated coded schemes [6], at the University of Stanford by Goldsmith and her team, studying the effects of variable-rate, variable-power arrangements [10] and at Southampton University in the UK, investigating a variety of practical aspects of AQAM [12, 13]. The channel's quality is estimated on a burst-by-burst basis and the most appropriate modulation mode is selected in order to maintain the required target bit error rate (BER) performance, whilst maximizing the system's Bit Per Symbol (BPS) throughput. Using this reconfiguration regime the distribution of channel errors becomes typically less bursty, than in conjunction with non-adaptive modems, which potentially increases the channel coding gains. Furthermore, the soft-decision channel codec metrics can be also invoked in estimating the instantaneous channel quality, irrespective of the type of channel impairments.

A range of coded AQAM schemes were analysed by Matsuoka *et al* [6], Lau *et al* [14] and Goldsmith *et al* [10]. For data transmission systems, which do not necessarily require a low transmission delay, variable–throughput adaptive schemes can be devised, which operate efficiently in conjunction with powerful error correction codecs, such as long block length turbo codes. However, the acceptable turbo interleaving delay is rather low in the context of low-delay interactive speech. Video communications systems typically require a higher bitrate than speech systems and hence they can afford a higher interleaving delay.

The above principles - which were typically investigated in the context of narrowband modems - were further advanced in conjunction with wideband modems, employing powerful block turbo coded wideband Decision Feedback Equaliser (DFE) assisted AQAM transceivers [15]. A neural-network Radial Basis Function (RBF) DFE based AQAM modem design was proposed in [16], where the RBF DFE provided the channel quality estimates for the modem mode switching regime. This modem was capable of removing the residual BER of conventional DFEs, when linearly non-separable received phasor constellations were encountered.

The above burst-by-burst adaptive principles can also be extended to Adaptive Orthogonal Frequency Division Multiplexing (AOFDM) schemes [17]. The associated AQAM principles were invoked in the context of parallel AOFDM modems also by Czylwik et al [18], Fischer [19] and Chow et al [20].

Our main contribution is that upon invoking the technique advocated - irrespective of the channel conditions experienced - the transceiver achieves always the best possible video quality by automatically

adjusting the achievable bitrate and the associated video quality in order to match the channel quality experienced. This is achieved on a near-instantaneous basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile outdoor propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

3.2 ACDMA Signalling Scenarios

ACDMA transmission parameter adaptation is an action of the transmitter in response to time-varying channel conditions. It is only suitable for duplex communication between two stations, since the transmission parameter adaptation relies on some form of channel estimation and signalling. In order to efficiently react to the changes in channel quality, the following steps have to be taken:

- *Channel quality estimation:* In order to appropriately select the transmission parameters to be employed for the next transmission, a reliable prediction of the channel quality during the next active transmit timeslot is necessary.
- *Choice of the appropriate parameters for the next transmission:* Based on the prediction of the expected channel conditions during the next timeslot, the transmitter has to select the appropriate modulation schemes for the subcarriers.
- *Signalling or blind detection of the employed parameters:* The receiver has to be informed, as to which set of demodulator parameters to employ for the received packet. This information can either be conveyed within the packet, at the cost of loss of useful data bandwidth, or the receiver can attempt to estimate the parameters employed at the transmitter by means of blind detection mechanisms.

Depending on the channel characteristics, these operations can be performed at either of the duplex stations, as shown in Figures 1(a), 1(b) and 1(c). If the channel is reciprocal, then the channel quality estimation for each link can be extracted from the reverse link, and we refer to this regime as open–loop adaptation. In this case, the transmitter needs to communicate the transmission parameter set to the receiver (Figure 1(a)), or the receiver can attempt blind detection of the transmission parameters employed (Figure 1(c)).

If the channel is not reciprocal, then the channel quality estimation has to be performed at the receiver of the link. In this case, the channel quality measure or the set of requested transmission parameters is

communicated to the transmitter in the reverse link (Figure 1(b)). This mode is referred to as closed–loop adaptation.

3.3 Video Transceiver

The schematic of the whole system is depicted in Figure 2. The multimedia source signal generated by the video encoder of Figure 2 is assembled into transmission packets constituting a CDMA transmission burst and the bits may be additionally mapped by the Mapper of Figure 2 to *n* number of different Forward Error Correction (FEC) protection classes. These bits are then convenveyed to the optional Time Division Multiplex (TDMA)/ Time Division Duplex (TDD) scheme of Figure 2, before they are assigned to the AQAM/ACDMA modem seen in Figure 2.

Again, the philosophy of the proposed burst-by-burst adaptive joint detection CDMA scheme is that the signal to interference plus noise ratio (SINR) at the output of the multi-user receiver is used in order to estimate the instantaneous channel quality. In one of its possible embodiments the receiver then decides on the transmitter's mode to be used during the next transmission burst on the basis of the received signal quality and the receiver's perception of the channel quality is signalled to the remote transmitter, in order to allow it to satisfy the receiver's integrity requirement.

In this study we transmitted 176x144 pixel Quarter Common Intermediate Format (QCIF) and 128x96 pixel Sub-QCIF (SQCIF) video sequences at 10 frames/s using a reconfigurable Time Division Multiple Access / Code Division Multiple Access (TDMA / CDMA) transceiver, which can be configured as a 1, 2 or 4 bit/symbol scheme shown in Figure 2. The H.263 video codec exhibits an impressive compression ratio, although this is achieved at the cost of a high vulnerability to transmission errors, since a run-length coded stream is rendered undecodable by a single bit error. In order to mitigate this problem, when the channel codec protecting the video stream is overwhelmed by the transmission errors, we refrain from decoding the corrupted video packet in order to prevent error propagation through the reconstructed video frame buffer [2]. We found that it was more beneficial in video quality terms, if these corrupted video packets were dropped and the reconstructed frame buffer was not updated, until the next video packet replenishing the specific video frame area was received. The associated video performance degradation was found perceptually unobjectionable for packet dropping- or transmission frame error rates (FER) below about 5%. These packet dropping events were signalled to the remote decoder by superimposing a strongly protected one-bit packet acknowledgement flag on the reverse-direction packet, as outlined in [2]. Bose-Chaudhuri-Hocquenghem (BCH) and turbo error correction codes were used and again, the CDMA transceiver was capable of transmitting 1, 2 and 4 bits per symbol, where each symbol was spread using a low spreading factor (SF) of 16, as seen in Table 1.

Parameter	
Multiple access	TDMA/CDMA
Channel type	COST 207 Bad Urban
Number of paths in channel	7
Normalised Doppler frequency	3.7×10^{-5}
CDMA spreading factor	16
Spreading sequence	Random
Frame duration	4.615 ms
Burst duration	577 μs
Joint detection CDMA receiver	Whitening matched filter (WMF) or Minimum mean square error block decision feedback equalizer (MMSE-BDFE)
No. of Slots/Frame	8
TDMA frame length	4.615ms
TDMA slot length	577μs
TDMA slots/Video packet	3
Chip Periods/TDMA slot	1250
Data Symbols/TDMA slot	68
User Data Symbol Rate (kBd)	14.7
System Data Symbol Rate (kBd)	117.9

Table 1: Generic system parameters using the Frames spread speech/data mode 2 proposal [11]

The associated parameters will be addressed in more depth during our further discourse. Employing a low spreading factor of 16 allowed us to improve the system's multi-user performance with the aid of joint-detection techniques [21]. We note furthermore that the implementation of the joint detection receivers is independent of the number of bits per symbol associated with the modulation mode used, since the receiver simply inverts the associated system matrix and invokes a decision concerning the received symbol, irrespective of how many bits per symbol were used. **Therefore, joint detection receivers are amenable to amalgamation with the above 1, 2 and 4 bit/symbol modem, since they do not have to be reconfigured each time the modulation mode is switched.**

In this performance study we used the Pan-European FRAMES proposal [11] as the basis for our CDMA system. The associated transmission frame structure is shown in Figure 3, while a range of generic system

Features	BCH coding	Turbo coding
Modulation	4QAM	
Transmission bitrate (kbit/s)	29.5	
Video-rate (kbit/s)	13.7	11.1
Video framerate (Hz)	10	

Table 2: FEC-protected and unprotected BCH and Turbo coded bitrates for the 4QAM transceiver mode

parameters are summarised in Table 1. In our performance studies we used the COST207 seven-path bad urban (BU) channel model, whose impulse response is portrayed in Figure 4.

Our initial experiments compared the performance of a whitening matched filter (WMF) for single user detection and the Minimum mean square error block decision feedback equalizer (MMSE-BDFE) for joint multi-user detection. These simulations were performed using 4-level Quadrature Amplitude Modulation (4QAM), transmitting both binary BCH and turbo coded video packets. The associated bitrates are summarised in Table 2.

The transmission bitrate of the 4QAM modem mode was 29.5Kbps, which was reduced due to the approximately half-rate BCH or turbo coding, plus the associated video packet acknowledgement feedback flag error control and video packetisation overhead to produce effective video bitrates of 13.7Kbps and 11.1Kbps, respectively. A more detailed discussion on the video packet acknowledgement feedback error control and video packetisation overhead will be provided in Section 3.4 with reference to the convolutionally coded multi-mode investigations.

Figure 5 portrays the bit error ratio (BER) performance of the BCH coded video transceiver using both matched filtering and joint detection for 2–8 users. The bit error ratio is shown to increase, as the number of users increases, even upon employing the MMSE-BDFE multi-user detector (MUD). However, while the matched filtering receiver exhibits an unacceptably high BER for supporting perceptually unimpaired video communications, the MUD exhibits a far superior BER performance.

When the BCH codec was replaced by the turbo-codec, the bit error ratio performance of both matched filtering and the MUD receiver improved, as shown in Figure 6. However, as expected, matched filtering was still outperformed by the joint detection scheme for the same number of users. Furthermore, the matched filtering performance degraded rapidly for more than two users.

Figure 7 shows the video packet loss ratio (PLR) for the turbo coded video stream using matched filtering and joint detection for 2–8 users. The figure clearly shows that the matched filter was only capable of meeting the target packet loss ratio of 5% for upto four users, when the channel SNR was in excess of 11dB. However, the joint detection algorithm guaranteed the required video packet loss ratio performance

Features	Multi-rate System		
Mode	BPSK	4QAM	16QAM
Bits/Symbol	1	2	4
FEC	Convolutional Coding		
Transmitted bits/packet	204	408	816
Total bitrate (kbit/s)	14.7	29.5	58.9
FEC-coded bits/packet	102	204	408
Assigned to FEC-coding (kbit/s)	7.4	14.7	29.5
Error detection per packet	16 bit CRC		
Feedback bits / packet	9		
Video packet size	77	179	383
Packet header bits	8	9	10
Video bits/packet	69	170	373
Unprotected video-rate (kbit/s)	5.0	12.3	26.9
Video framerate (Hz)	10		

Table 3: Operational-mode specific transceiver parameters for the proposed multi-mode system

for 2–8 users in the entire range of channel SNRs shown. Furthermore, the 2-user matched-filtered PLR performance was close to the 8-user MUD PLR.

3.4 Multi-mode Video System Performance

Having shown that joint detection can substantially improve our system's performance, we investigated the performance of a multi-mode convolutionally coded video system employing joint detection, while supporting two users. The associated convolutional codec parameters are summarised in Table 3. Below we now detail the video packetisation method employed. The reader is reminded that the number of symbols per TDMA frame was 68 according to Table 1. In the 4QAM mode this would give 136 bits per TDMA frame. However, if we transmitted one video packet per TDMA frame, then the packetisation overhead would absorb a large percentage of the available bitrate. Hence we assembled larger video packets, thereby reducing the packetisation overhead and arranged for transmitting the contents of a video packet over three consecutive TDMA frames, as indicated in Table 1. Therefore each protected video packet consists of $68 \times 3 = 204$ modulation symbols, yielding a transmission bitrate of between 14.7 and 38.9 Kbps for BPSK and 16QAM, respectively. However, in order to protect the video data

we employed half-rate, constraint-length nine convolutional coding, using octal generator polynomials of 561 and 753. The useful video bitrate was further reduced due to the 16-bit Cyclic Redundancy Checking (CRC) used for error detection and the nine-bit repetition-coded feedback error flag for the reverse link. This results in video packet sizes of 77, 179 and 383 bits for each of the three modulation modes. The useful video capacity was finally further reduced by the video packet header of between 8 and 10 bits, resulting in useful or effective video bitrates ranging from 5 to 26.9 Kbps in the BPSK and 16QAM modes, respectively.

The proposed multi-mode system can switch amongst the 1, 2 and 4 bit/symbol modulation schemes under network control, based upon the prevailing channel conditions. As seen in Table 3, when the channel is benign, the unprotected video bitrate will be approximately 26.9Kbps the 16QAM mode. However, as the channel quality degrades, the modem will switch to the BPSK mode of operation, where the video bitrate drops to 5Kbps, and for maintaining a reasonable video quality, the video resolution has to be reduced to SQCIF (128x96 pels).

Figure 8 portrays the packet loss ratio for the multi-mode system, in each of its modulation modes for a range of channel SNRs. It can be seen in the figure that above a channel SNR of 14dB the 16QAM mode offers an acceptable packet loss ratio of less than 5%, while providing an unprotected video rate of about 26.9Kbps. If the channel SNR drops below 14dB, the multi-mode system is switched to 4QAM and eventually to BPSK, when the channel SNR is below 9dB, in order to maintain the required quality of service, which is dictated by the packet loss ratio. The figure also shows the acknowledgement feedback error ratio (FBER) for a range of channel SNRs, which has to be substantially lower, than the video PLR itself. This requirement is satisfied in the figure, since the feedback errors only occur at extremely low channel SNRs, where the packet loss ratio is approximately 50%, and it is therefore assumed that the multi-mode system would have switched to a more robust modulation mode, before the feedback acknowledgement flag can become corrupted.

The video quality is commonly measured in terms of the peak-signal-to-noise-ratio (PSNR). Figure 9 shows the video quality in terms of the PSNR versus the channel SNRs for each of the modulation modes. As expected, the higher throughput bitrate of the 16QAM mode provides a better video quality. However, as the channel quality degrades, the video quality of the 16QAM mode is reduced and hence it becomes beneficial to switch from the 16QAM mode to 4QAM at an SNR of about 14dB, as it was suggested by the packet loss ratio performance of Figure 8. Although the video quality expressed in terms of PSNR is superior for the 16QAM mode in comparison to the 4QAM mode at channel SNRs in excess of 12dB, however, due to the excessive PLR the perceived video quality appears inferior in comparison to that of the 4QAM mode, even though the 16QAM PSNR is higher for channel SNRs

in the range of 12–14dB. More specifically, we found that it was beneficial to switch to a more robust modulation scheme, when the PSNR was reduced by about 1dB with respect to its unimpaired PSNR value. This ensured that the packet losses did not become subjectively apparent, resulting in a higher perceived video quality and smoother degradation, as the channel quality deteriorated.

The effect of packet losses on the video quality quantified in terms of PSNR is portrayed in Figure 10. The figure shows, how the video quality degrades, as the PLR increases. It has been found that in order to ensure a seamless degradation of video quality as the channel SNR reduced, it was the best policy to switch to a more robust modulation scheme, when the PLR exceeded 5%. The figure clearly shows that a 5% packet loss ratio results in a loss of PSNR, when switching to a more robust modulation scheme. However, if the system did not switch until the PSNR of the more robust modulation mode was similar, the perceived video quality associated with the originally higher rate, but channel-impaired stream became inferior.

3.5 Burst-by-Burst adaptive videophone system

A burst-by-burst adaptive modem, maximizes the system's throughtput by using the most appropriate modulation mode for the current instantaneous channel conditions. Figure 11 exemplifies, how a burst-by-burst adaptive modem changes its modulation modes based on the fluctuating channel conditions. The adaptive modem uses the SINR estimate at the output of the joint-detector to estimate the instantaneous channel quality, and hence to set the modulation mode.

The probability of the adaptive modem using each modulation mode for a particular channel SNRs is portrayed in Figure 12. It can be seen at high channel SNRs that the modem mainly uses the 16QAM modulation mode, while at low channel SNRs the BPSK mode is most prevalent.

The advantage of dynamically reconfigured burst-by-adaptive modem over the statically switched multi-mode system previously described, is that the video quality is smoothly degraded as the channel conditions deteriorate. The switched multi-mode system results in more sudden reductions in video quality, when the modem switches to a more robust modulation mode. Figure 13 shows the throughput bitrate of the dynamically reconfigured burst-by-burst adaptive modem, compared to the three modes of the statically switched multi-mode system. The reduction of the fixed modem modes' effective throughput at low SNRs is due to the fact that under such channel conditions an increased fraction of the transmitted packets have to be dropped, reducing the effective throughtput. The figure shows the smooth reduction of the throughput bitrate, as the channel quality deteriorates. The burst-by-burst modem matches the BPSK mode's bitrate at low channel SNRs, and the 16QAM mode's bitrate at high SNRs. The dynamically reconfigured burst-by-burst adaptive modem characterised in the figure perfectly estimates the prevalent

channel conditions although in practice the estimate of channel quality is not perfect and it is inherently delayed. However, we have found that non-perfect channel estimates result in only slightly reduced performance, when compared to perfect channel estimation.

The smoothly varying throughput bitrate of the burst-by-burst adaptive modem translates into a smoothly varying video quality as the channel conditions change. The video quality measured in terms of the average peak signal to noise ratio (PSNR) is shown versus the channel SNR in Figure 14 in contrast to that of the individual modem modes. The figure demonstrates that the burst-by-burst adaptive modem provides equal or better video quality over a large proportion of the SNR range shown than the individual modes. However, even at channel SNRs, where the adaptive modem has a slightly reduced PSNR, the perceived video quality of the adaptive modem is better since the video packet loss rate is far lower, than that of the fixed modem modes.

Figure 15 shows the video packet loss ratio versus channel SNR for the three fixed modulation modes and the burst-by-burst adaptive modem with perfect channel estimation. Again the figure demonstrates that the video packet loss ratio of the adaptive modem is similar to that of the fixed BPSK modem mode, however the adaptive modem has a far higher bitrate throughput, as the channel SNR increases. The burst-by-burst adaptive modem gives an error performance similar to that of the BPSK mode, but with the flexibity to increase the bitrate throughput of the modem, when the channel conditions improve. If imperfect channel estimation is used, the throughput bitrate of the adaptive modem is reduced slightly. Furthermore, the video packet loss ratio seen in Figure 15 is slightly higher for the AQAM scheme due to invoking higher-order modem modes, as the channel quality increases. However we have found that is possible to maintain the video packet loss ratio within tolerable limits for the range of channel SNRs considered.

The interaction between the video quality measured in terms of PSNR and the video packet loss ratio can be more clearly seen in Figure 16. The figure shows that the adaptive modem slowly degrades the decoded video quality from that of the error free 16QAM fixed modulation mode, as the channel conditions deteriorate. The video quality degrades from the error-free 41dB PSNR, while maintaining a near-zero video packet loss ratio, until the PSNR drops below about 36dB PSNR. At this point the further reduced channel quality inflicts an increased video packet loss rate and the video quality degrades more slowly. The PSNR versus packet loss ratio performance then tends toward that achieved by the fixed BPSK modulation mode. However the adaptive modem achieved better video quality than the fixed BPSK modem even at high packet loss rates.

3.6 Summary

A joint-detection assisted multimode CDMA-based video transceiver was proposed, which substantially outperformed the conventional matched-filtering based transceiver, which was characterised by adaptively reconfiguring the transceiver's mode of operation based on the instantaneous channel quality. In our transceiver a higher number of bits per modulation symbol was invoked by the transmitter, when the channel quality was sufficiently high for supporting this more bitrate efficient, but less error resilient transmission mode. By contrast, a more error resilient but less bitrate efficient mode was invoked for supporting error-free CDMA transmission over wireless multi-user channels.

In this embodiment the above property was exploited in a practical adaptive video transceiver, which instructed the video codec to generate the required number of bits that the CDMA transceiver was capable of delivering in its current channel-condition dependent configuration mode.

In other embodiments the proposed burst-by-burst adaptive transceiver can be invoked in the context of arbitrary multimedia signals, irrespective of their resolution or source representation quality. Specific further embodiments of such codecs are constituted by programmable-rate speech, audio, video, handwriting codecs, which can be configured by the transceiver to generate a channel-quality dependent number of source-coded bits.

The proposed burst-by-burst adaptive video transceiver guaranteed a near-unimpaired video quality for channel SNRs in excess of about 5 dB over the COST207 dispersive Rayleigh-faded channel. The benefits of the multimode video transceiver clearly manifest themselves in terms of supporting un-impaired video quality under time-variant channel conditions, where a single-mode transceiver's quality would become severely degraded by channel effects. The dynamically reconfigured burst-by-burst adaptive modem gave better perceived video quality due to its more graceful reduction in video quality, as the channel conditions degraded, than a statically switched multi-mode system.

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CLAIMS

1. A method for CDMA transmission of a multimedia signal over a transmission medium, the method comprising:

providing a transmitter operable to transmit in a plurality of modulation modes varying in bit rate and error resilience between a highest bit rate, lowest error resilience mode and a lowest bit rate, highest error resilience mode;

obtaining a channel quality measure for current transmission; and

switching to a more or less error resilient modulation mode each time the channel quality measure respectively degrades or improves by a defined amount, whereby multimedia signal quality varies smoothly with varying channel quality of the transmission medium.

2. A method according to claim 1, wherein the channel quality measure is a multimedia signal quality dependent signal-to-noise value.

3. A method according to claim 2, wherein the defined amount is set with reference to an unimpaired signal-to-noise value.

4. A method according to claim 2 or 3, wherein the signal-to-noise value is a peak signal-to-noise ratio for a multimedia video signal.

5. A method according to claim 2 or 3, wherein the signal-to-noise value is a segmental signal-to-noise ratio for a multimedia speech signal.

6. A method according to claim 1, wherein the channel quality measure is a packet loss value.

7. A method according to claim 1, wherein the packet loss value is varied dependent upon desired multimedia signal quality.

8. A method according to claim 1, wherein the channel quality measure is a bit error rate.

9. A method according to any one of claims 1 to 8, wherein the channel quality measure is based on monitoring signal integrity at a remote receiver.

10. A method according to any one of claims 1 to 8, wherein the channel quality measure is based on monitoring signal integrity at a receiver local to the transmitter.

11. A transmitter for transmission of a multimedia source signal over a transmission medium to a remote receiver, the transmitter comprising a CDMA modem having an output for transmitting a multimedia source signal and an input for receiving a channel quality measure for current transmission, wherein the CDMA modem is switchable between a plurality of modulation modes varying in bit rate and error resilience between a highest bit rate, lowest error resilience mode and a lowest bit rate, highest error resilience mode, such that the CDMA modem is switched to a more or less error resilient modulation mode each time the channel quality measure respectively degrades or improves by a defined amount, whereby multimedia signal quality is smoothly variable with varying channel quality of the transmission medium.

12. A transmitter according to claim 11, wherein the channel quality measure is a multimedia signal quality dependent signal-to-noise value.

13. A transmitter according to claim 12, wherein the defined amount is set with reference to an unimpaired signal-to-noise value.

14. A transmitter according to claim 12 or 13, wherein the signal-to-noise value is a peak signal-to-noise ratio for a multimedia video signal.

15. A transmitter according to claim 12 or 13, wherein the signal-to-noise value is a segmental signal-to-noise ratio for a multimedia speech signal.

16. A transmitter according to claim 11, wherein the channel quality measure is a packet loss value.

17. A transmitter according to claim 11, wherein the packet loss value is variable dependent upon desired multimedia signal quality.

18. A transmitter according to claim 11, wherein the channel quality measure is a bit error rate.

19. A transmitter according to any one of claims 11 to 18, wherein the channel quality measure is based on monitoring signal integrity at a remote receiver.

20. A transmitter according to any one of claims 11 to 18, wherein the channel quality measure is based on monitoring signal integrity at a receiver local to the transmitter.

21. A transmission system for transmission of multimedia source signals over a transmission medium, the system comprising:

a first transceiver including a local receiver and a local transmitter according to any one of claims 11 to 20; and

a second transceiver including a remote receiver and a remote transmitter according to any one of claims 11 to 20.

Fig. 1(a)

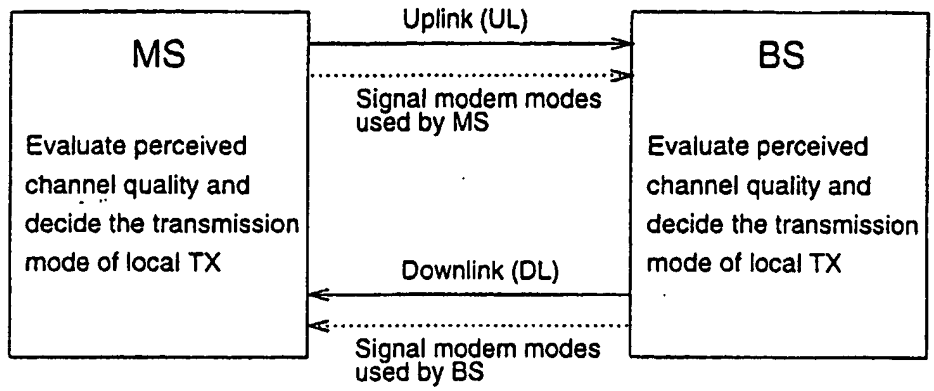


Fig. 1(b)

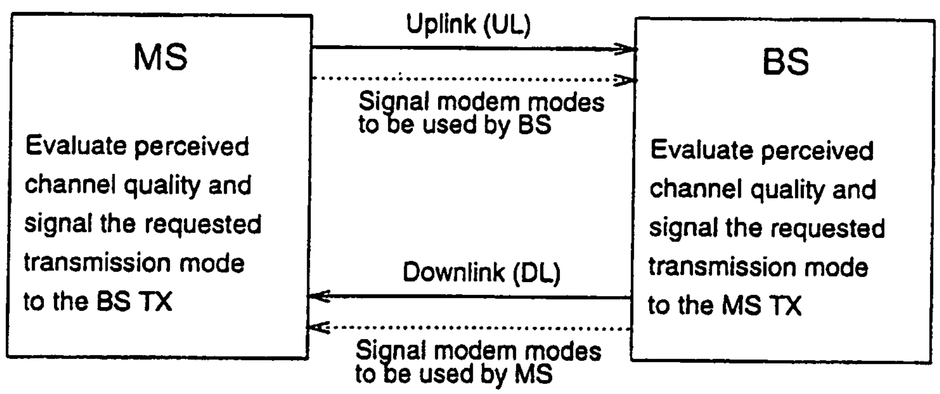


Fig. 1(c)

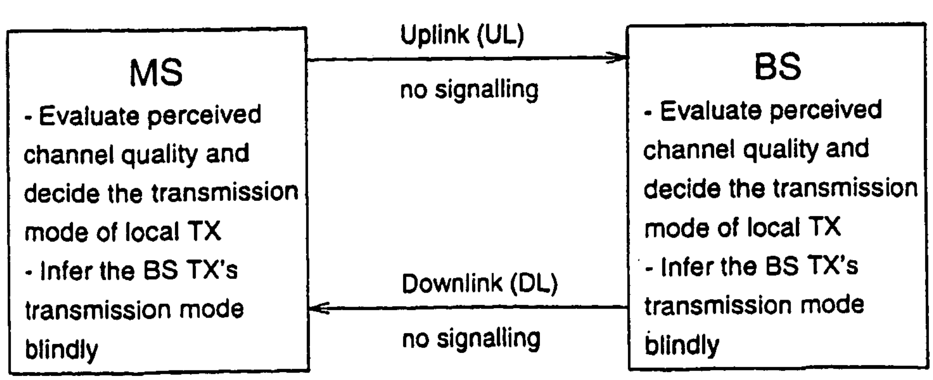


Fig. 2

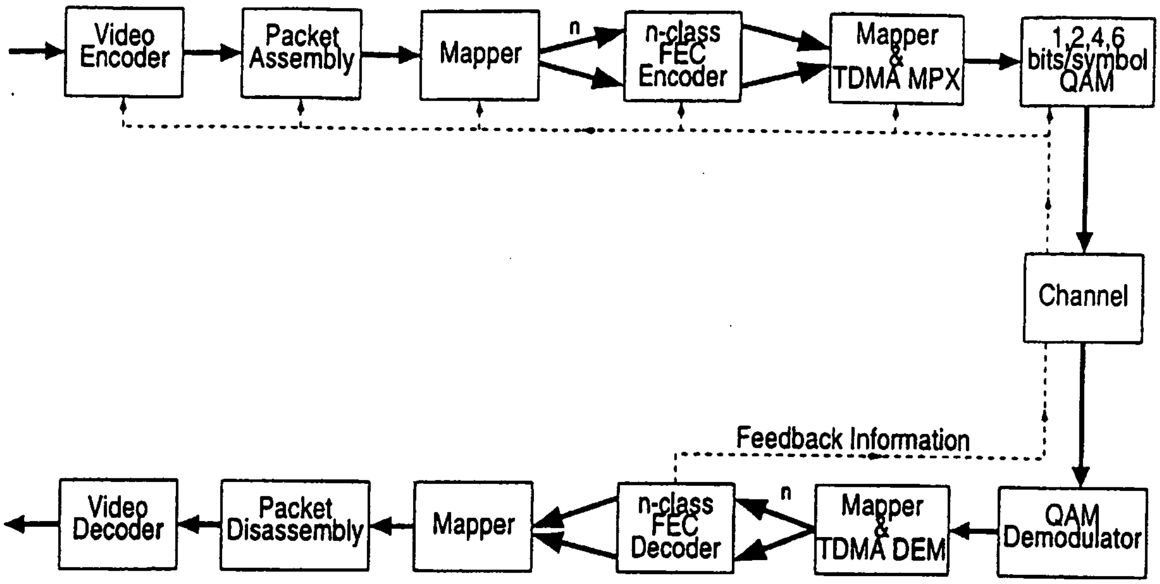
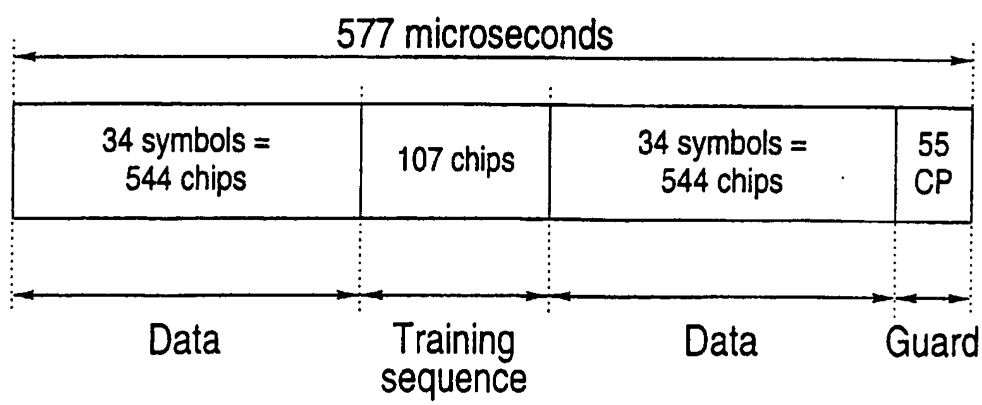
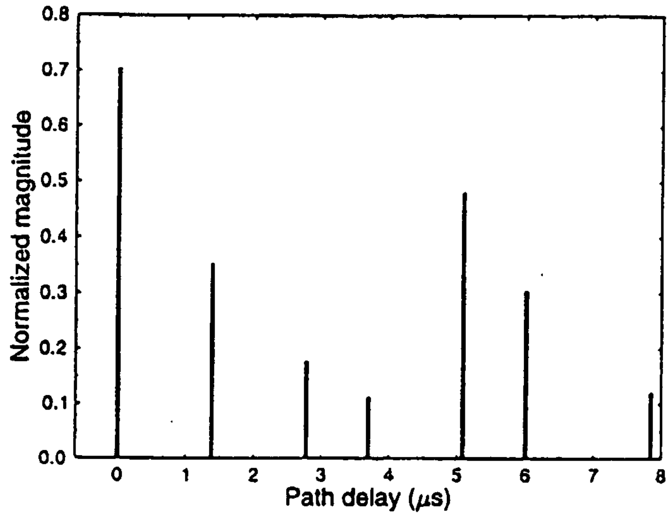


Fig. 3



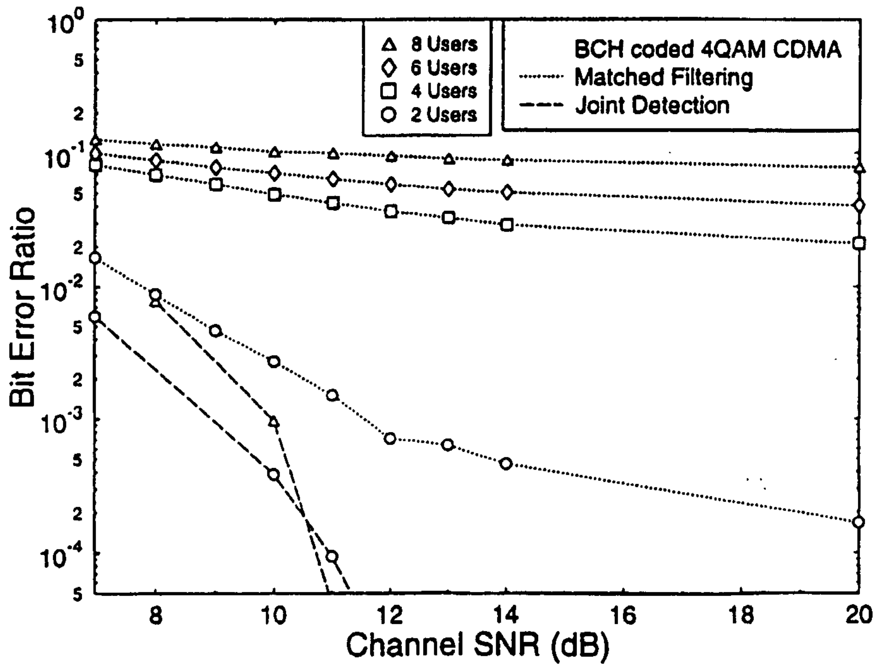
Spread speech/data burst 2

Fig. 4



Normalized magnitude
Path delay (μs)

Fig. 5



8 Users
6 Users
4 Users
2 Users
BCH coded 4QAM CDMA
Matched Filtering
Joint Detection
Bit Error Ratio
Channel SNR (dB)

Fig. 6

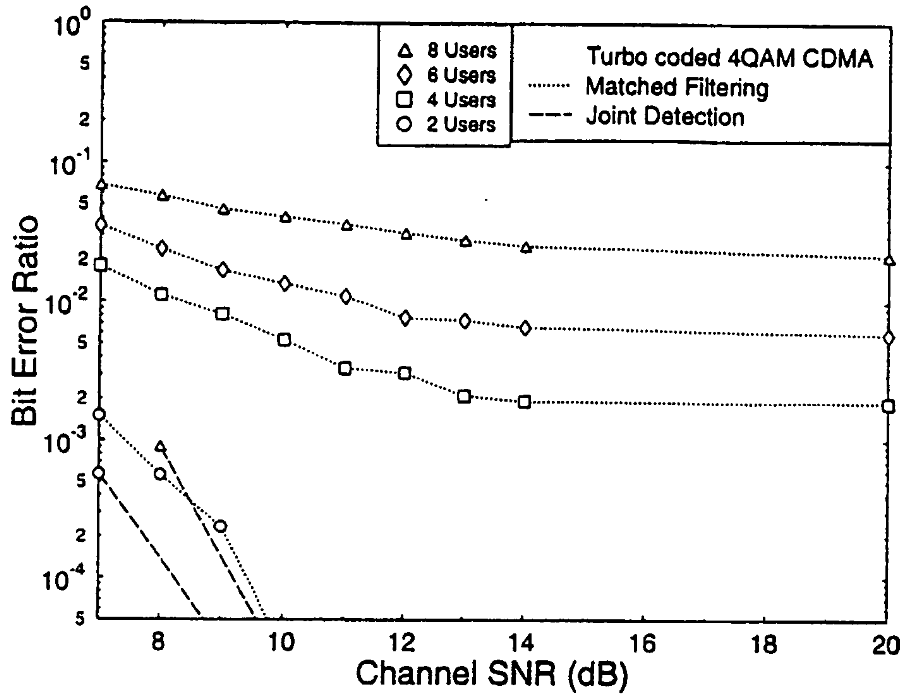

Bit Error Ratio
Channel SNR (dB)
8 Users
6 Users
4 Users
2 Users
Turbo coded 4QAM CDMA
Matched Filtering
Joint Detection

Fig. 7

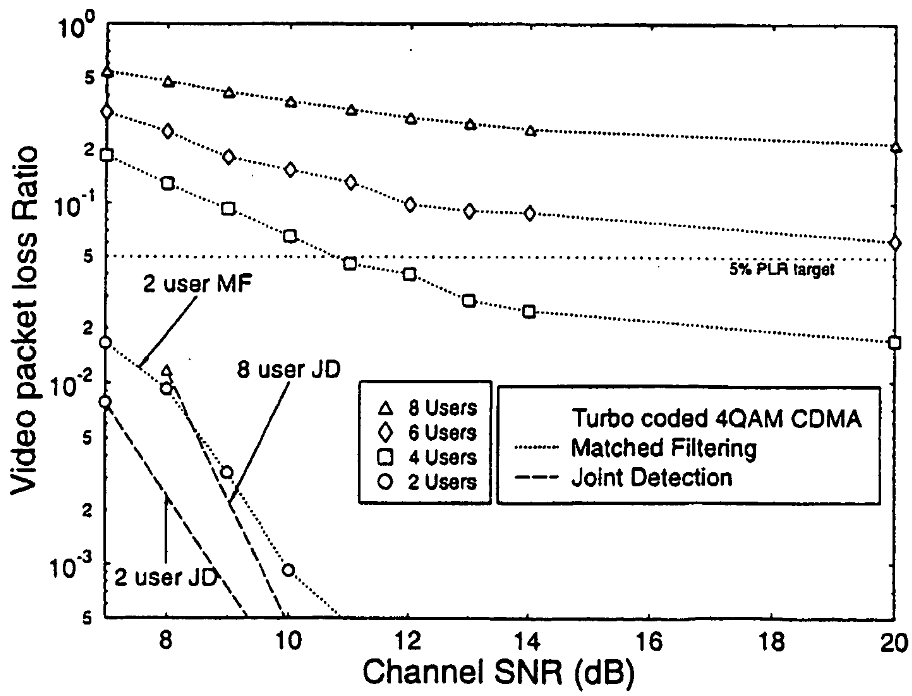

Video packet loss Ratio
Channel SNR (dB)
5% PLR target
2 user MF
8 user JD
2 user JD
8 Users
6 Users
4 Users
2 Users
Turbo coded 4QAM CDMA
Matched Filtering
Joint Detection

Fig. 8

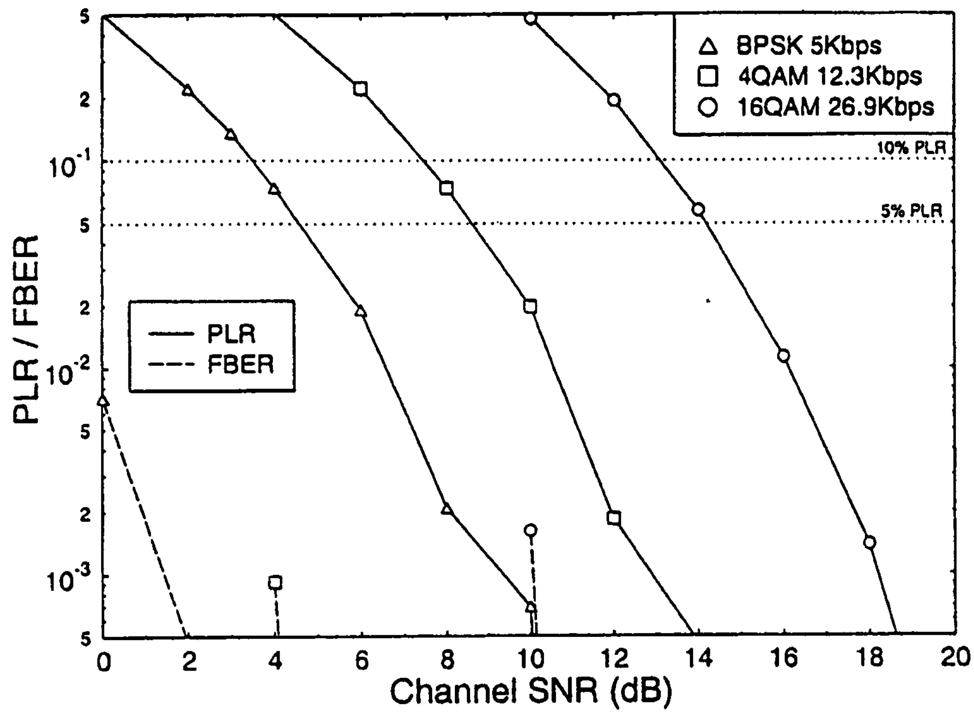

BPSK 5Kbps
4QAM 12.3Kbps
16QAM 26.9Kbps
10% PLR
5% PLR
PLR
FBER
PLR / FBER
Channel SNR (dB)

Fig. 9

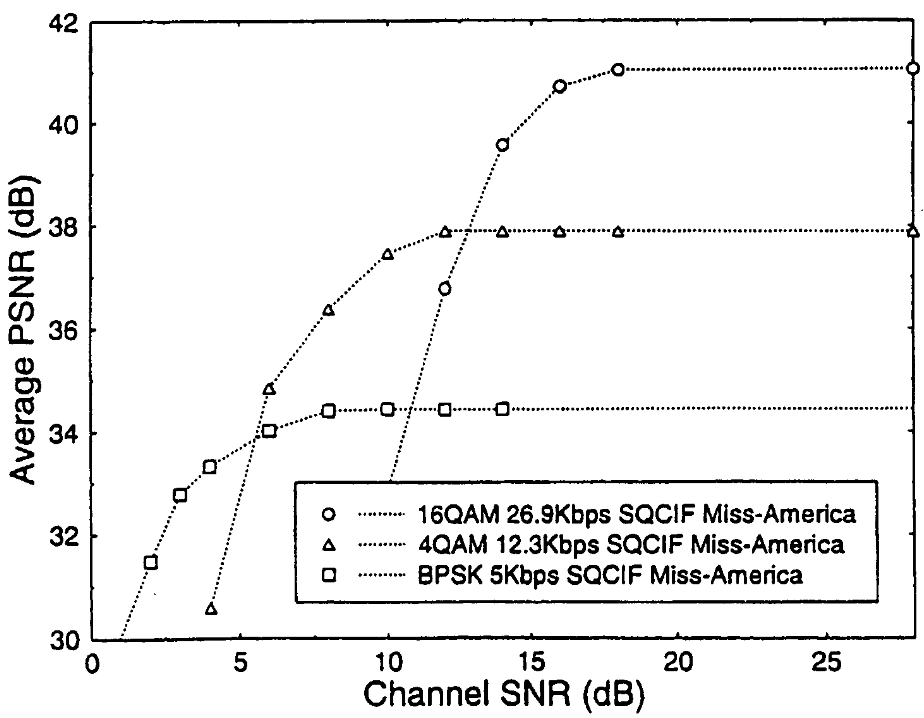

16QAM 26.9Kbps SQCIF Miss-America
4QAM 12.3Kbps SQCIF Miss-America
BPSK 5Kbps SQCIF Miss-America
Average PSNR (dB)
Channel SNR (dB)

Fig. 10

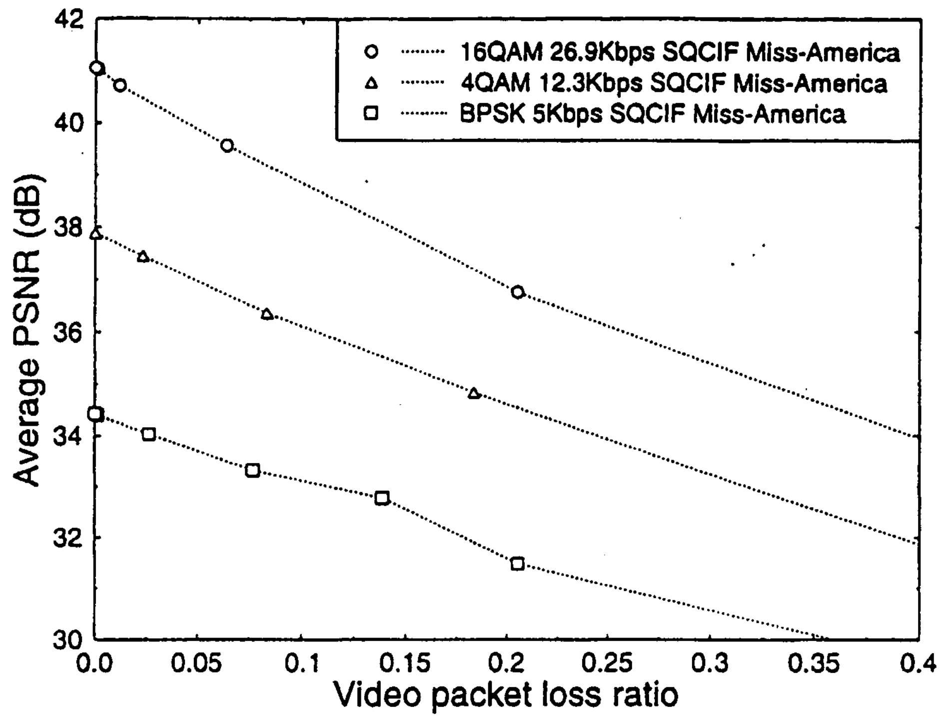

16QAM 26.9Kbps SQCIF Miss-America
4QAM 12.3Kbps SQCIF Miss-America
BPSK 5Kbps SQCIF Miss-America
Average PSNR (dB)
Video packet loss ratio
42
40
38
36
34
32
30
0.0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4

Fig. 11

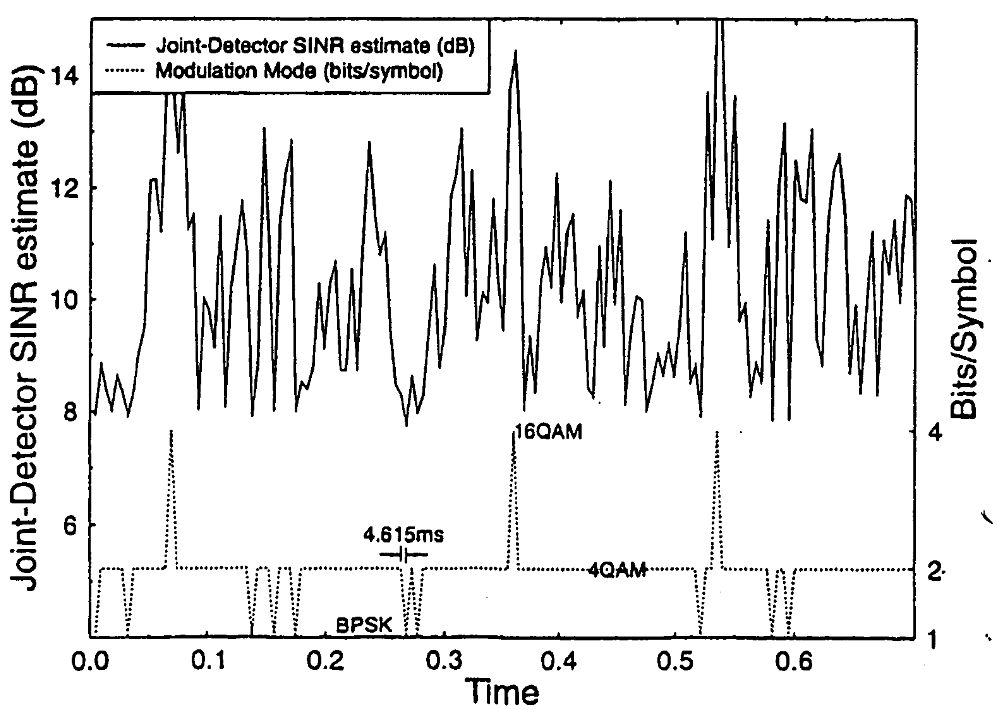

Joint-Detector SINR estimate (dB)
Modulation Mode (bits/symbol)
Joint-Detector SINR estimate (dB)
Bits/Symbol
Time
16QAM
4QAM
BPSK
4.615ms
14
12
10
8
6
4
2
1
0.0
0.1
0.2
0.3
0.4
0.5
0.6

Fig. 12

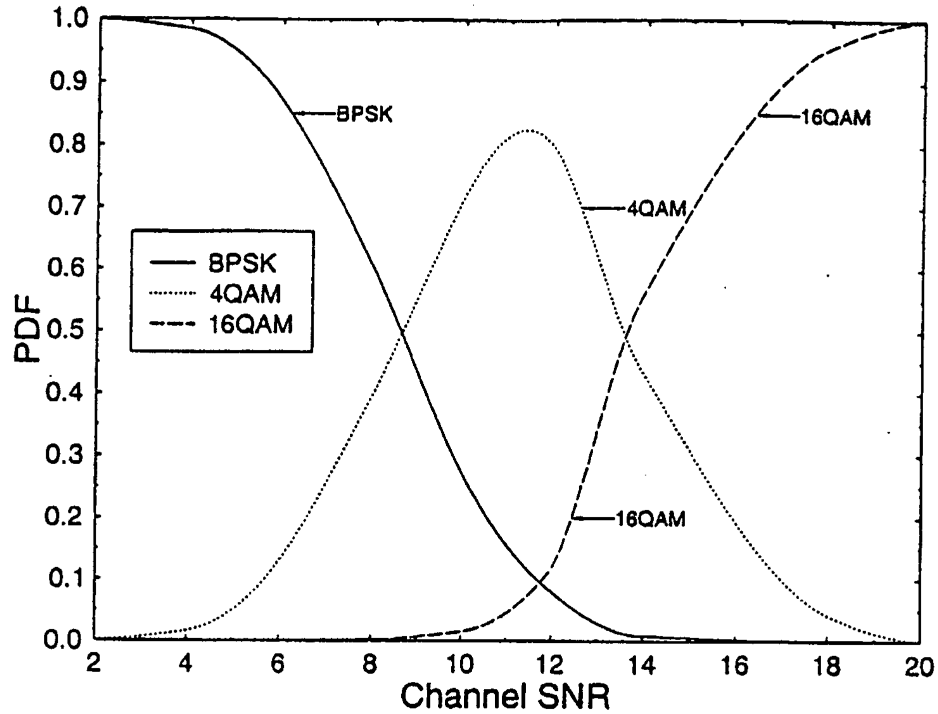

1.0
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0.0
PDF
2
4
6
8
10
12
14
16
18
20
Channel SNR
BPSK
4QAM
16QAM
BPSK
4QAM
16QAM
16QAM

Fig. 13

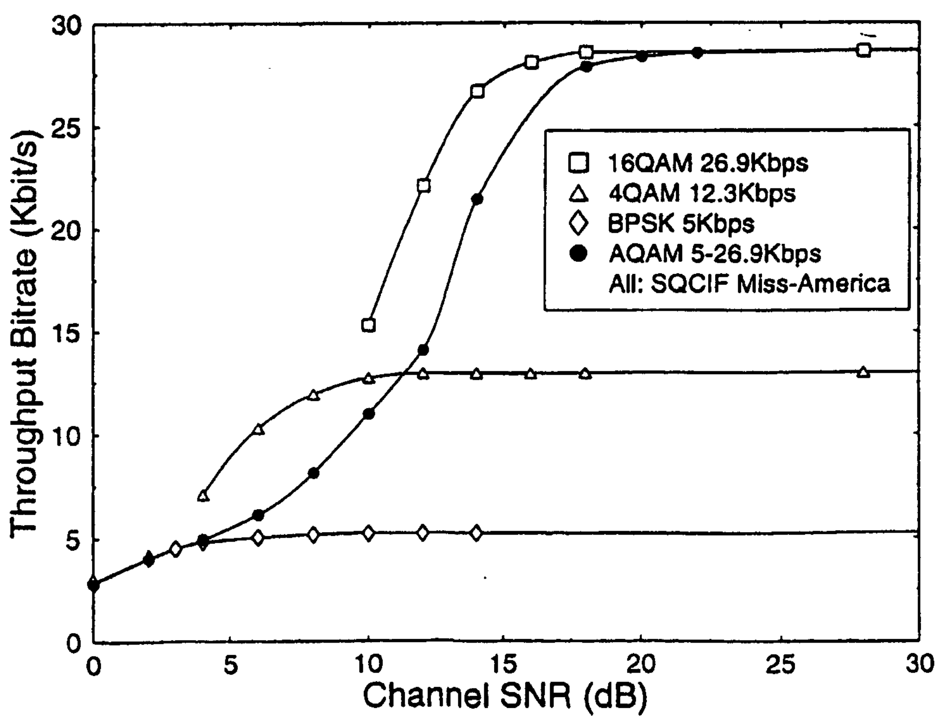

30
25
20
15
10
5
0
Throughput Bitrate (Kbit/s)
0
5
10
15
20
25
30
Channel SNR (dB)
16QAM 26.9Kbps
4QAM 12.3Kbps
BPSK 5Kbps
AQAM 5-26.9Kbps
All: SQCIF Miss-America

Fig. 14

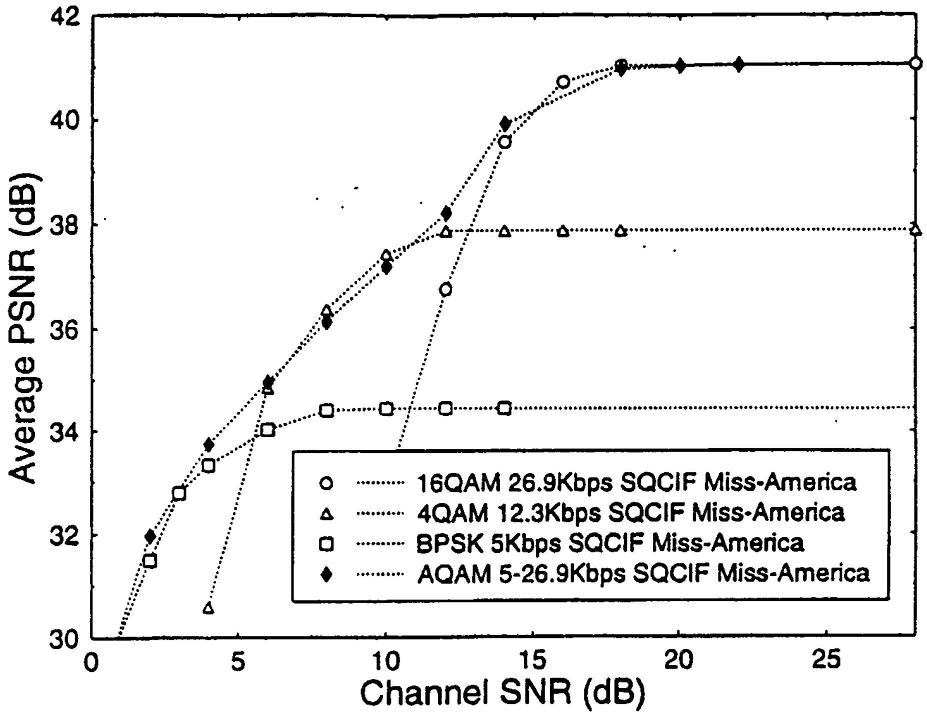

42
40
38
36
34
32
30
Average PSNR (dB)
0
5
10
15
20
25
Channel SNR (dB)
16QAM 26.9Kbps SQCIF Miss-America
4QAM 12.3Kbps SQCIF Miss-America
BPSK 5Kbps SQCIF Miss-America
AQAM 5-26.9Kbps SQCIF Miss-America

Fig. 15

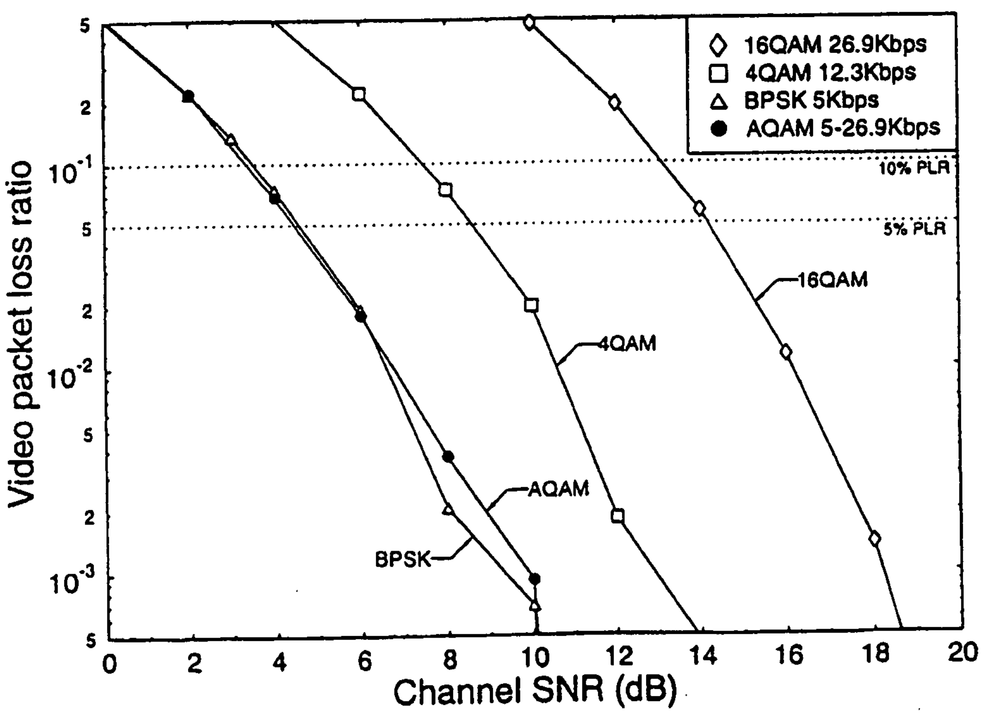
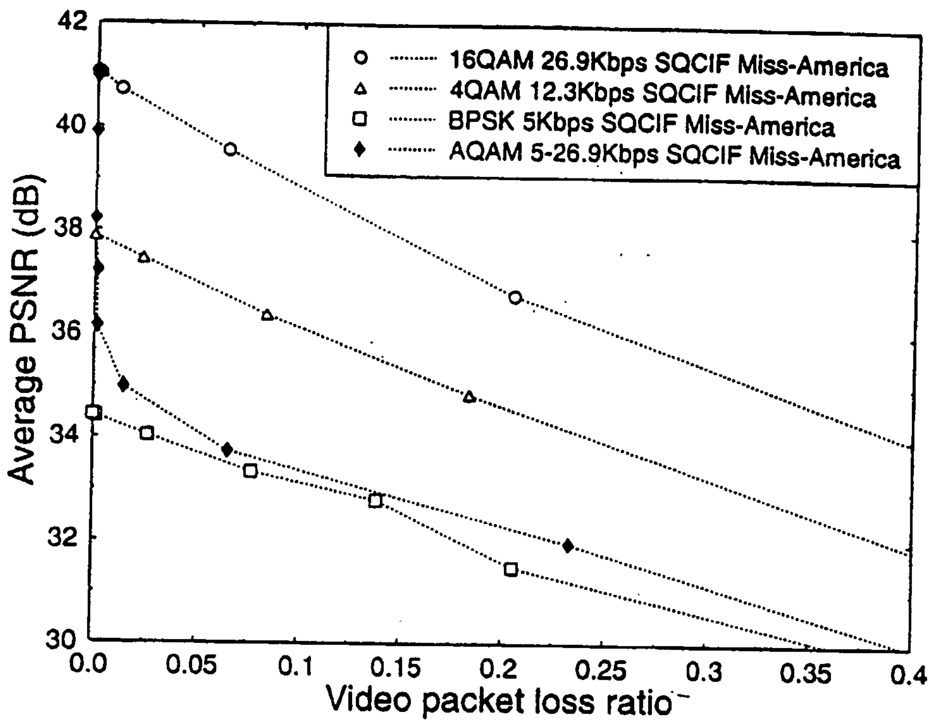

Video packet loss ratio
10^{-1}
10^{-2}
10^{-3}
10% PLR
5% PLR
16QAM 26.9Kbps
4QAM 12.3Kbps
BPSK 5Kbps
AQAM 5-26.9Kbps
16QAM
4QAM
AQAM
BPSK
0
2
4
6
8
10
12
14
16
18
20
Channel SNR (dB)

Fig. 16


16QAM 26.9Kbps SQCIF Miss-America
4QAM 12.3Kbps SQCIF Miss-America
BPSK 5Kbps SQCIF Miss-America
AQAM 5-26.9Kbps SQCIF Miss-America
Average PSNR (dB)
42
40
38
36
34
32
30
0.0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
Video packet loss ratio